the post-transform processor processing the transform coefficient sequence to thereby generate first and second transform coefficient sequences by combining and/or differencing first and second selected transform coefficients from the fast Fourier transform coefficient sequence, the post-transform processor further processing each of the first and second transform coefficient sequences to thereby generate audio coded frequency domain coefficients so as to generate respective sequences of said audio coded frequency domain coefficients for the first and second audio channels.

A5 cont

38. (New) The apparatus of claim 37 wherein the pre-transform processor determines a transform length for each of the channels and pairs the channels according to their determined transform length, the coding the audio samples of first and second channels in each pair being performed based on the determined transform length.

39. (New) A apparatus of claim 28, further comprising applying a windowing function in combination with the pre-multiplication factor.

REMARKS

Claims 1-39 are now present in this application. Many of the previously entered claims have been amended to remove multiple claim dependencies and to further clarify the invention. New apparatus claims 28-39 are added.

The Applicants kindly request examination on the claims now presented. If questions arise regarding the application, the Examiner is invited to contact the undersigned at (206) 622-4900.

Respectfully submitted,

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VERSION WITH MARKINGS TO SHOW CHANGES MADE

In the Claims:

Claims 3-10, 12-17, 19-24, 26 and 27 have been amended as follows:

3. (Amended) A method for coding audio data as claimed in claim 1~~ or 2~~, wherein the first trigonometric function factor for each audio sample is a function of the audio sample sequence position and the number of samples in the sequence.

4. (Amended) A method for coding audio data as claimed in claim 1~~3~~, wherein the respective second trigonometric function factors for each transform coefficient in the sequence are respective functions of the transform coefficient sequence position and the number of coefficients in the sequence.

5. (Amended) A method for coding audio data as claimed in claim 1~~4~~, wherein the respective third trigonometric function factors are respective functions of the transform coefficient sequence position.

6. (Amended) A method for coding audio data as claimed in claim 1~~5~~, wherein step i) comprises multiplying the input sequence samples $x[n]$ by the first trigonometric function factor $cos(\pi n/N)$ to generate the intermediate sample sequence, where:

$x[n]$ are the input sequence audio samples;

N is the number of input sequence audio samples; and

$n = 0,\dots,N-1$.

7. (Amended) A method for coding audio data as claimed in claim 1~~6~~, wherein step ii) comprises computing the fast Fourier transform of the intermediate sample sequence so as to generate said transform coefficient sequence $G_k = g_{k,r} + jg_{k,i}$, where:

G_k is the transform coefficient sequence;

$g_{k,r}$ are the real transform coefficient components;

$g_{k,i}$ are the imaginary transform coefficient components; and

$k = 0,\ldots,(N/2-1)$.

8. (Amended) A method for coding audio data as claimed in claim 1~~7~~, wherein step iii) comprises determining the addition stream coefficients T_2 and subtraction stream coefficients T_1 according to:

$$T_1 = g_{k,r} \cos(\pi(k+1/2)/N) - g_{k,i} \sin(\pi(k+1/2)/N)$$

$$T_2 = g_{k,r} \cos(\pi(k+1/2)/N) + g_{k,i} \sin(\pi(k+1/2)/N)$$

where T_1 and T_2 are the subtraction stream and addition stream coefficients, respectively.

9. (Amended) A method for coding audio data as claimed in claim 1~~8~~, wherein steps iv) and v) comprise generating the audio coded frequency domain coefficients X_k according to:

$$X_k = T_1 \cos(\pi(2k+1)/4) - T_2 \sin(\pi(2k+1)/4)$$

where Xk are the audio coded frequency domain coefficients; and

$\cos(\pi(2k+1)/4)$ and $\sin(\pi(2k+1)/4)$ are the third trigonometric function factors.

10. (Amended) A method for coding audio data, including the steps of:

combining first and second sequences of digital audio samples from first and second audio channels into a single complex sample sequence;

determining a Fourier transform coefficient sequence~~ as defined in any preceding claim~~;

generating first and second transform coefficient sequences by combining and/or differencing first and second selected transform coefficients from said Fourier transform coefficient sequence; and

for each of the first and second transform coefficient sequences, generating audio coded frequency domain coefficients ~~as defined in any preceding claim, so as~~ to generate respective sequences of said audio coded frequency domain coefficients for the first and second audio channels.

12. (Amended) A method for coding audio data as claimed in claim 10~~ or 11~~, wherein the complex sample sequence is processed by multiplying ~~step i) comprises multiplying~~ the input sequence samples $z[n]$ by ~~the~~ a first trigonometric function factor $cos(\pi n/N) + jsin(\pi n/N)$ to generate ~~the~~ an intermediate sample sequence, where:

$z[n] = x[n] + jy[n]$ is the complex sample sequence;

$x[n]$ is the first sequence of digital audio samples;

$y[n]$ is the second sequence of digital audio samples;

N is the number of input sequence audio samples in each sequence;

$n = 0,....,N-1$; and

j is the complex constant.

13. (Amended) A method for coding audio data as claimed in claim 11~~ or 12~~, wherein said first and second transform coefficient sequences are generated according to:

$$G_k = (Z_k + Z^*_{N-k-1})/2$$

$$G'_k = (Z_k - Z^*_{N-k-1})/2j$$

where G_k is said first transform coefficient sequence;

G'_k is said second transform coefficient sequence;

N is the number of input sequence audio samples;

$k = 0,....,(N/2-1);$

Z_k is said first transform coefficient;

Z^*_{N-k-1} is the complex conjugate of said second transform coefficient; and

j is the complex constant.

14. (Amended) A method for coding audio data as claimed in claim 10 ~~any one of claims 10 to 13, including~~ further comprising examining said first and second sequences of digital audio samples to determine a short or long transform length, and coding the audio samples using a short or long transform length as determined.

15. (Amended) A method for coding audio data comprising sequences of digital audio samples from a plurality of audio channels as defined in claim 10, further

comprising determining a transform length for each of the channels, pairing the channels according to their determined transform length, and coding the audio samples of first and second channels in each pair~~, as defined in any one of claims 10 to 13,~~ according to the determined transform length.

16. (Amended) A method for coding audio data as claimed in ~~any preceding~~ claim 10, including applying a windowing function in combination with ~~said~~ multiplying ~~step i)~~ the complex sample sequence by a first trigonometric function factor.

17. (Amended) A method for coding audio data including the steps of:

obtaining at least one input sequence of digital audio samples;

pre-processing the input sequence samples including applying a pre-multiplication factor to obtain modified input sequence samples;

transforming the modified input sequence samples into a transform coefficient sequence ~~utilising~~utilizing a fast Fourier transform; and

post-processing the sequence of transform coefficients including applying first post-multiplication factors to the real and imaginary coefficient components, differencing and combining the post-multiplied real and imaginary components, applying second post-multiplication factors to the difference and combination results, and differencing to obtain a sequence of modified discrete cosine transform coefficients representing said input sequence of digital audio samples.

19. (Amended) A method as claimed in claim ~~18~~17, wherein the pre-multiplication factor applied to each digital audio sample in the input sequence is a trigonometric function of the audio sample sequence position and the number of samples in the sequence.

20. (Amended) A method as claimed in claim ~~18~~17, wherein the first post-multiplication factors for each transform coefficient in the sequence are trigonometric

functions of the transform coefficient sequence position and the number of coefficients in the sequence.

21. (Amended) A method as claimed in claim ~~18~~17, wherein the second post-multiplication factor for each difference or combination result is trigonometric functions of the transform coefficient sequence position of the coefficients used in the difference or combination.

22. (Amended) A method as claimed in claim 17~~any one of claims 17 to 21~~, wherein the pre-processing operations are performed on each sample in the input sequence individually.

23. (Amended) A method as claimed in claim 17~~any one of claims 17 to 22~~, wherein the post-processing operations are performed on each transform coefficient in the sequence individually.

24. (Amended) A method for coding audio data including the steps of:

obtaining first and second input sequences of digital audio samples corresponding to respective first and second audio channels;

combining the first and second input sequences of digital audio samples into a single complex input sample sequence;

pre-processing the complex input sequence samples including applying a pre-multiplication factor to obtain modified complex input sequence samples;

transforming the modified complex input sequence samples into a complex transform coefficient sequence ~~utilising~~utilizing a fast Fourier transform; and

post-processing the sequence of complex transform coefficients to obtain first and second sequences of audio coded frequency domain coefficients corresponding to the first and second audio channels including, for each corresponding frequency domain coefficient in the first and second sequences, selecting first and second complex transform coefficients from said sequence of complex transform coefficients, combining the first complex transform coefficient and the complex conjugate of the second complex transform coefficient for said first channel and differencing the first complex transform coefficient and the complex conjugate of the second complex transform coefficient for said second channel, and applying respective post-multiplication factors to the combination and difference to obtain said audio coded frequency domain coefficients corresponding to the first and second audio channels.

26. (Amended) A method as claimed in claim 24~~ or 25~~, wherein the post-processing for each of the first and second channels includes applying first post-multiplication factors to the real and imaginary coefficient components, differencing and combining the post-multiplied real and imaginary components, applying second post-multiplication factors to the difference and combination results, and differencing to obtain a sequence of modified discrete cosine transform coefficients representing said input sequence of digital audio samples.

27. (Amended) A method for coding audio data including the steps of:

obtaining first and second input sequences of digital audio samples $x[n]$, $y[n]$ corresponding to respective first and second audio channels;

combining the first and second input sequences of digital audio samples into a single complex input sample sequence $z[n]$, where $z[n] = x[n] + jy[n]$;

pre-processing the complex input sequence samples including applying a pre-multiplication factor $cos(\pi n/N) + jsin(\pi n/N)$ to obtain modified complex input sequence samples, where N is the number of audio samples in each of the first and second input sequences and $n = 0,....,(N-1)$;

transforming the modified complex input sequence samples into a complex transform coefficient sequence Z_k ~~utilising~~utilizing a fast Fourier transform, wherein $k = 0,....,(N12- 1)$; and

post-processing the sequence of complex transform coefficients to obtain first and second sequences of audio coded frequency domain coefficients corresponding to the first and second audio channels X_k, Y_k according to:

$$G_k = (Z_k + Z^*_{N-k-1})/2 \qquad k=0...N/2-1$$
$$G'_k = (Z_k - Z^*_{N-k-1})/2j \qquad k=0...N/2-1$$

$$X_k = \cos\gamma * (g_{k,r}\cos(\pi(k+1/2)/N) - g_{k,i}\sin(\pi(k+1/2)/N)$$
$$- \sin\gamma * (g_{k,r}\sin(\pi(k+1/2)/N) + g_{k,i}\cos(\pi(k+1/2)/N)$$

$$Y_k = \cos\gamma * (g'_{k,r}\cos(\pi(k+1/2)/N) - g'_{k,i}\sin(\pi(k+1/2)/N)$$
$$- \sin\gamma * (g'_{k,r}\sin(\pi(k+1/2)/N) + g'_{k,i}\cos(\pi(k+1/2)/N)$$

where G_k is a transform coefficient sequence for the first channel;

G'_k is a transform coefficient sequence for the second channel;

$g_{k,r}$ and $g_{k,i}$ are the real and imaginary transform coefficient components of G_k;

$g'_{k,r}$ and $g'_{k,i}$ are the real and imaginary transform coefficient components of G'_k;

Z^*_{N-k-1} is the complex conjugate of Z_{N-k-1}; and

$\gamma(k) = \pi(2k+1)/4$.

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